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~~Informal Status Report~~

~~on~~

ON-BOARD PROCESSING FOR DATA-COLLECTION SATELLITE SYSTEMS

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SECTION 1: INTRODUCTION

1.1 PURPOSE

The purpose of this informal status report is three-fold:

a) To summarize the research effort that has been expended on this project in relation to the specific tasks that were embodied in the original research proposal.

b) To document this research effort by presenting major results and the conclusions that can be drawn from them.

c) To indicate those directions that seem most fruitful for continued investigation and to suggest certain tasks that might be abandoned in any follow-on effort.

The style of presentation is concise and limited to end-results without the preliminary analysis. The documentation consists of specific computer-generated curves selected from a much larger set of existing outputs. In discussing the conclusions a careful distinction is made between those that are tentative and those that are firm.

1.2 SUMMARY OF RESEARCH EFFORT

The specific tasks contained in the original research proposal are itemized, with some elaboration, in Table I along with an indication of the degree of effort that has been devoted to them and a summary of the major conclusions. These items are then expanded upon in Section 2.

Even a casual inspection of Table I makes it clear that the original research proposal was over-ambitious in terms of what could be accomplished by one or two inexperienced graduate students. This difficulty is easily overcome, of course, by extending the investigation over a longer time period.

TABLE I: SUMMARY OF PROPOSED TASKS

Proposed Task	Current Status	Location in This Report	Major Conclusions
<u>Coarse Frequency Measurement</u>			
Fast-Fourier Transform	Extensive analysis and computations completed. Variance of estimates determined.	2.2	Works for signals larger than -14 dB. May be too time-consuming.
Fast-Hadamard Transform	Analysis and preliminary computations for the noise-free case completed. Noise not yet considered.	2.3	May suppress weak signals and introduce false alarms because of sidelobes.
Digital Filter Bank	Preliminary design, no analysis.	2.4	Slower than FFT unless parallel processing is used.
Use of Hard Limiting	Discarded.	2.7.2	Not feasible
Implementation	Preliminary hardware designs for FFT and FHT.	2.7.1	None as yet
<u>Fine Frequency Measurement</u>			
Phase-lock Loops	Squaring loop and Costas loop considered. Analysis not complete.	2.5	No difficulty foreseen.
<u>Message Format Design</u>			
Optimum format	Mathematical model set up, not yet solved.	2.7.4	None as yet
Synchronization	Analysis begun, not completed.	2.7.4	None as yet
Preamble elimination	No effort	-	-

TABLE I (CONTINUED)

Proposed Task	Current Status	Location in This Report	Major Conclusions
Data Modulation and Demodulation	PSK assumed, no further effort.	2.7.3	No reason for using other than PSK.
System Error Analysis			
Frequency Identification	Probabilities of detection and false alarm determined for the FFT.	2.6	Acceptable values of P_D and P_F can be obtained with input SNR greater than -12 dB.
Data transmission errors	No effort	-	-
Data Reduction			
Software vs. Special Purpose Circuits	No effort	-	-
Time Scheduling	No effort	-	-
Coding for Re-transmission	No effort	-	-

However, a more serious criticism of the research effort to date is that it has been over-concerned with the investigation of some rather standard and prosaic estimation and detection procedures and has not really resolved the fundamental theoretical issues that must limit any practical solution. The concentration on standard transform methods that are computationally cumbersome because they ignore the available a priori information about the signal is an example of this. This aspect of the research effort, and some possible alternatives, are discussed in Section 3.

SECTION 2: DOCUMENTATION OF RESULTS

2.1 INTRODUCTION

The only topics discussed in detail in this section are those tasks from Table I for which there are some significant results. Other tasks that have not been carried to the point of reaching conclusions are noted briefly in a final sub-section, and tasks on which no effort has been expended are not considered at all.

The documentation consists of stating the objective of each task, presenting the pertinent equations, displaying typical curves (where appropriate) illustrating the results, and stating the major conclusions.

2.2 FAST FOURIER TRANSFORM (FFT) METHODS OF COARSE FREQUENCY MEASUREMENT

2.2.1 Objectives

An obvious method of deciding whether or not a short sample of observed data contains one or more sinusoids is to examine the frequency spectrum of that sample. If the duration, T, of the sample is great enough, and the noise is small enough, then separate peaks in the frequency spectrum will indicate the presence of sinusoids at (or near) the frequencies at which the peaks occur.

The degree to which two or more frequency components can be resolved depends upon both the sample duration, T, and variance of the estimate of the frequency spectrum. When the estimated spectrum is smoothed in order to reduce the variance, the sharpness of the peaks (and, hence, the resolution) is reduced. Thus, there is a trade-off between resolution and variance and there is an optimum degree of smoothing that should be used.

The ultimate objective in using the Fast Fourier transform in an investigation of coarse frequency measurements is to determine the optimum amount of

smoothing to use and thereby gain insight into the fundamental trade-offs. The reason for using Fourier methods, rather than methods that are computationally more efficient, is that the results are not obscured by the idiosycrasies of more specialized techniques that are not well understood. It is not anticipated that Fourier methods will be used in the final system realization because of the excessive amount of computation required.

2.2.2 Discussion of Results

When noise is present the Fourier transform, which is sensitive to phase, is not a good indicator of the frequency spectrum. The power spectrum, or spectral density, avoids the phase problem and exhibits greater statistical stability, particularly when smoothed appropriately. If the observed data in time T is represented by a set of N samples denoted by X_k, k = 0, 1, ..., N - 1, then the Fourier coefficient associated with a frequency of $f_n = n/T$ is given by

$$A_n = \sum_{k=0}^{N-1} X_k \, e^{-j \frac{2\pi kn}{N}} \quad , \quad n = 0, 1, \ldots, N-1 \tag{2-1}$$

and the corresponding power spectrum (as computed by the direct method) is

$$P(n/T) = \frac{T}{N} \, |A_n|^2 \tag{2-2}$$

The power spectrum can also be computed by the indirect method in which the autocorrelation function is first estimated from

$$R(nT/N) = \frac{1}{N-n} \sum_{k=0}^{N-n-1} X_k \, X_{k+n} \quad , \quad n = 0, 1, \ldots, N-1 \tag{2-3}$$

The power spectrum is then the Fourier cosine transform of the autocorrelation function. Thus

$$P(n/T) = \frac{T}{N}\left[R(0) + R(T)\cos(N-1)\pi + 2\sum_{k=1}^{N-2} R(kT/N)\cos\frac{kn\pi}{N} \right] \tag{2-4}$$

This spectrum is essentially equivalent to that of (2-2) but it does not have the same statistical properties. Although some work has been done on the indirect method, all of the results to be presented here were obtained by the direct method.

In order to smooth the spectral estimates of either (2-2) or (2-4) they must be convolved with an appropriate spectral window, $W(k)$. Thus, the smoothed estimate is

$$P_L(n/T) = \sum_{k=-L}^{L} W(n-k)\, P(k/T) \qquad n = 0, 1, \ldots, N/2 \qquad |L| \leq N/2 \tag{2-5}$$

Under suitable assumptions on the correlation between estimates and smoothness of the true spectrum, it can be shown that

$$E\left[P_L(n/T)\right] = \sum_{k=-L}^{L} W(k)\, E\left[P(n/T)\right] \tag{2-6}$$

$$\mathrm{Var}\left[P_L(n/T)\right] = \sum_{k=-L}^{L} W^2(k)\, \mathrm{Var}\left[P(n/T)\right] \tag{2-7}$$

This suggests that $\sum_{k=-L}^{L} W(k) = 1$ is the condition for an unbiased smoothed estimate when the raw estimate is unbiased. A measure of the reduction in variance of an unbiased estimate is given by

$$\frac{\xi}{2} = \frac{1}{\sum_{k=-L}^{L} W^2(k)} \tag{2-8}$$

where ξ is usually referred to as the degrees of freedom. A large value of

$\overset{\xi}{\cancel{\zeta}}$ is desirable in order to reduce the variance of the estimate as much as possible.

The equivalent spectral width of the estimate of a single frequency compenent is also proportional to $\overset{\xi}{\cancel{\zeta}}$. In particular, it has the form

$$B_e = \frac{\cancel{\zeta}\,\xi}{2T} \qquad \text{Hz} \tag{2-9}$$

Thus, any attempt to reduce the variance of the estimate will result in a corresponding increase in the spectral width of a single component. The objective, therefore, is to find the value of $\overset{\xi}{\cancel{\zeta}}$ that maximizes the probability of correctly detecting a given set of signal components. This problem is addressed further in Sec. 2.6.

An approximate indication of a reasonable value of $\overset{\xi}{\cancel{\zeta}}$ is obtained from the parameters of the physical situation. If it is desired to identify sinusoidal components having frequencies separated by 300 Hz, then B_e should be on the order of 300 Hz. If the sample length is T = 0.01, then from (2-9)

$$\zeta = 2B_eT = 2(300)(0.01) = 6 \tag{2-10}$$

A large number of power spectra for sinusoids with and without noise have been computed. Since there is no problem in correctly identifying the frequencies present in the noise-free case, the discussion here will concentrate on the noisy case only. The power spectra displayed in Figs. 1, 2, and 3 are typical of the results. In each of these there are two signals of equal amplitude at ~~4~~ 4.05 kHz and 23 kHz with varying amounts of noise. In Fig. 1 the signal-to-noise ratio (power in either signal to the total noise power in the 30 kHz band) is -10 dB for both spectra, but different noise samples are used for the top and bottom calculations. Fig. 2 presents the same situation for -14 dB signal-to-noise ratio and Fig. 3 repeats this for -15 dB. A noticeable degradation in detectability of the signal peaks occurs in this later case.

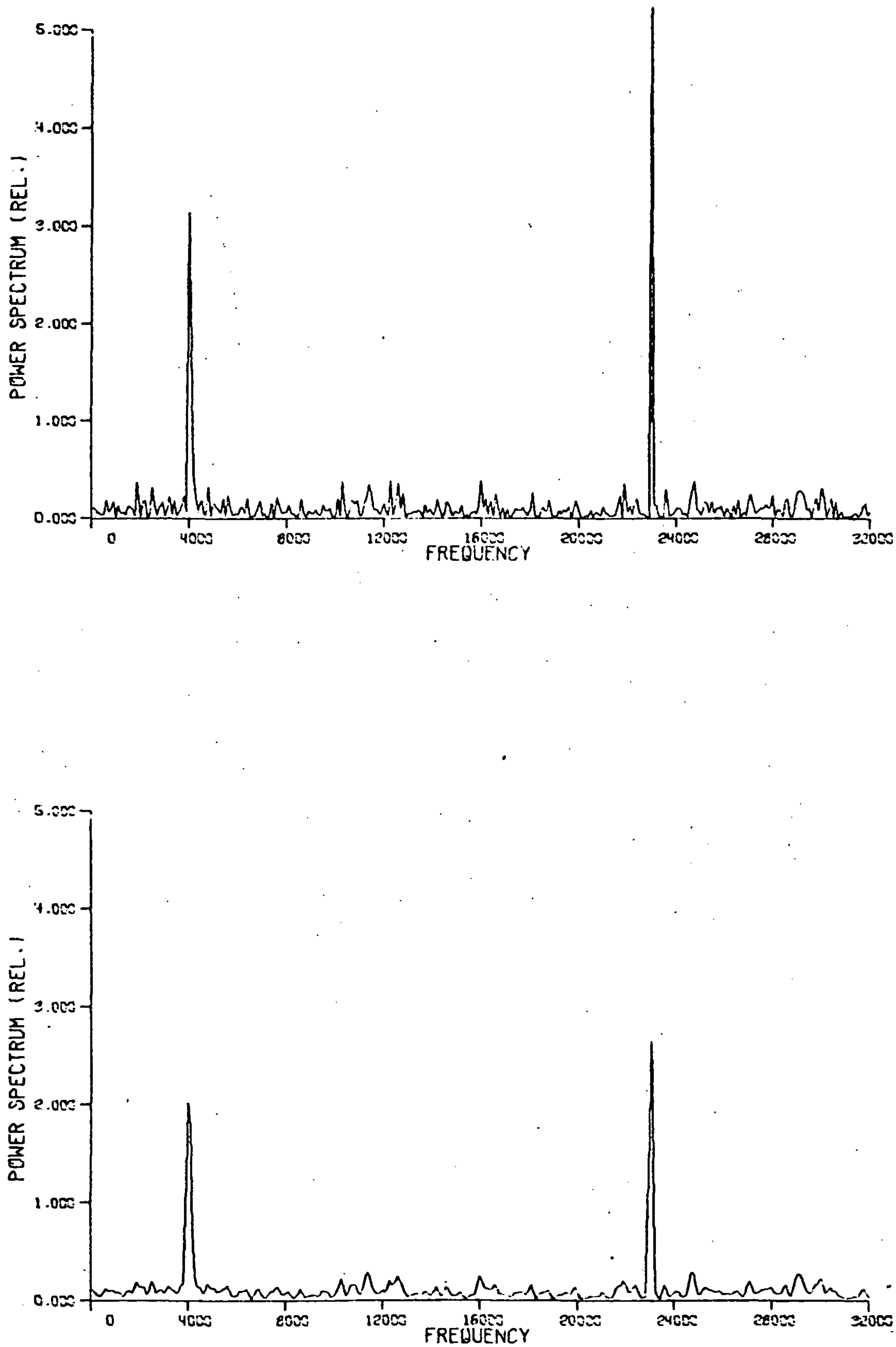

Fig. 1. *Power spectra when each signal is -10 dB below the total* noise. Upper and lower curves are for the same signals but different realizations of the noise, and the lower curve is smoothed.

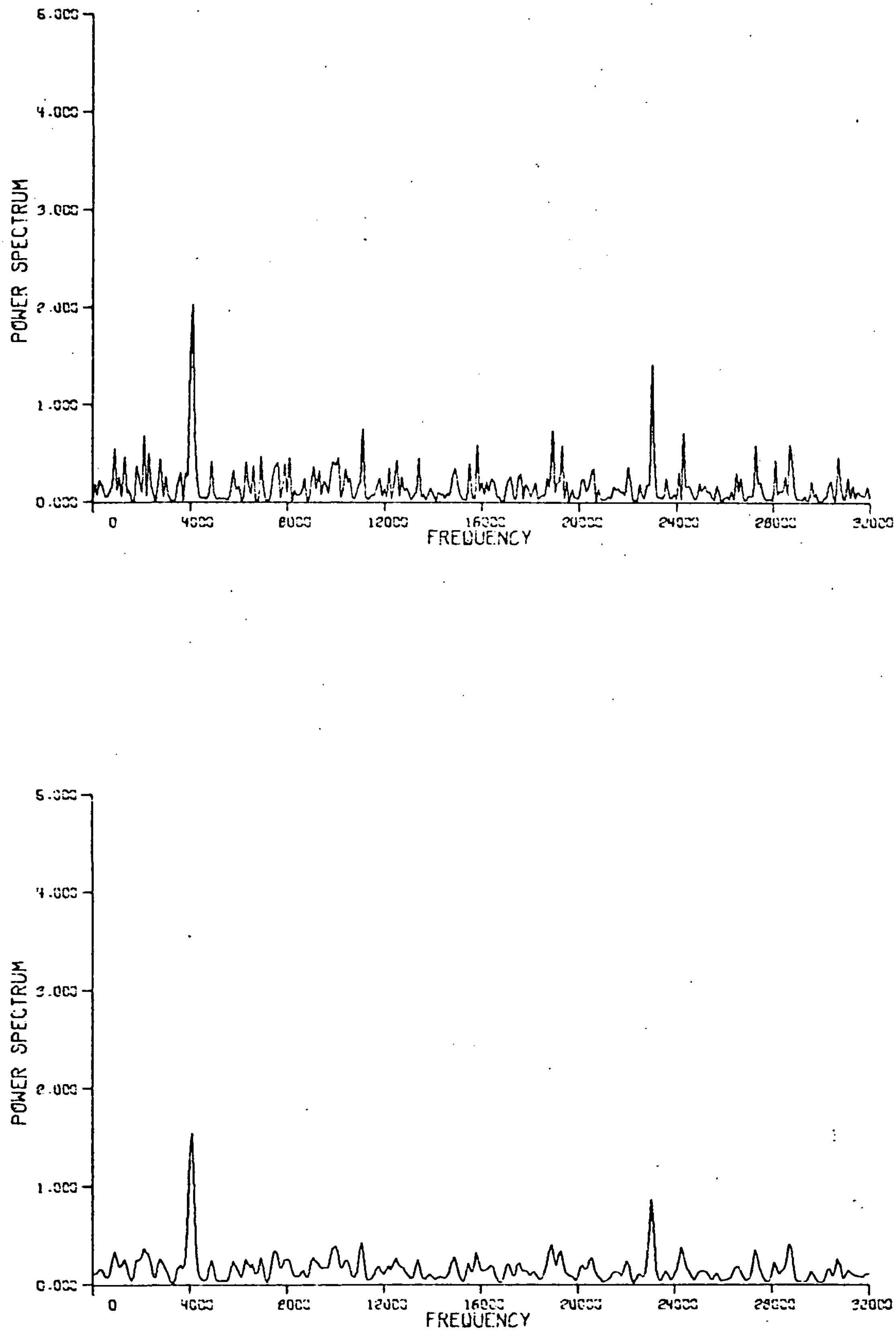

Fig. 2. Same signals as Fig. 1, but -14 dB signal-to-noise ratio.

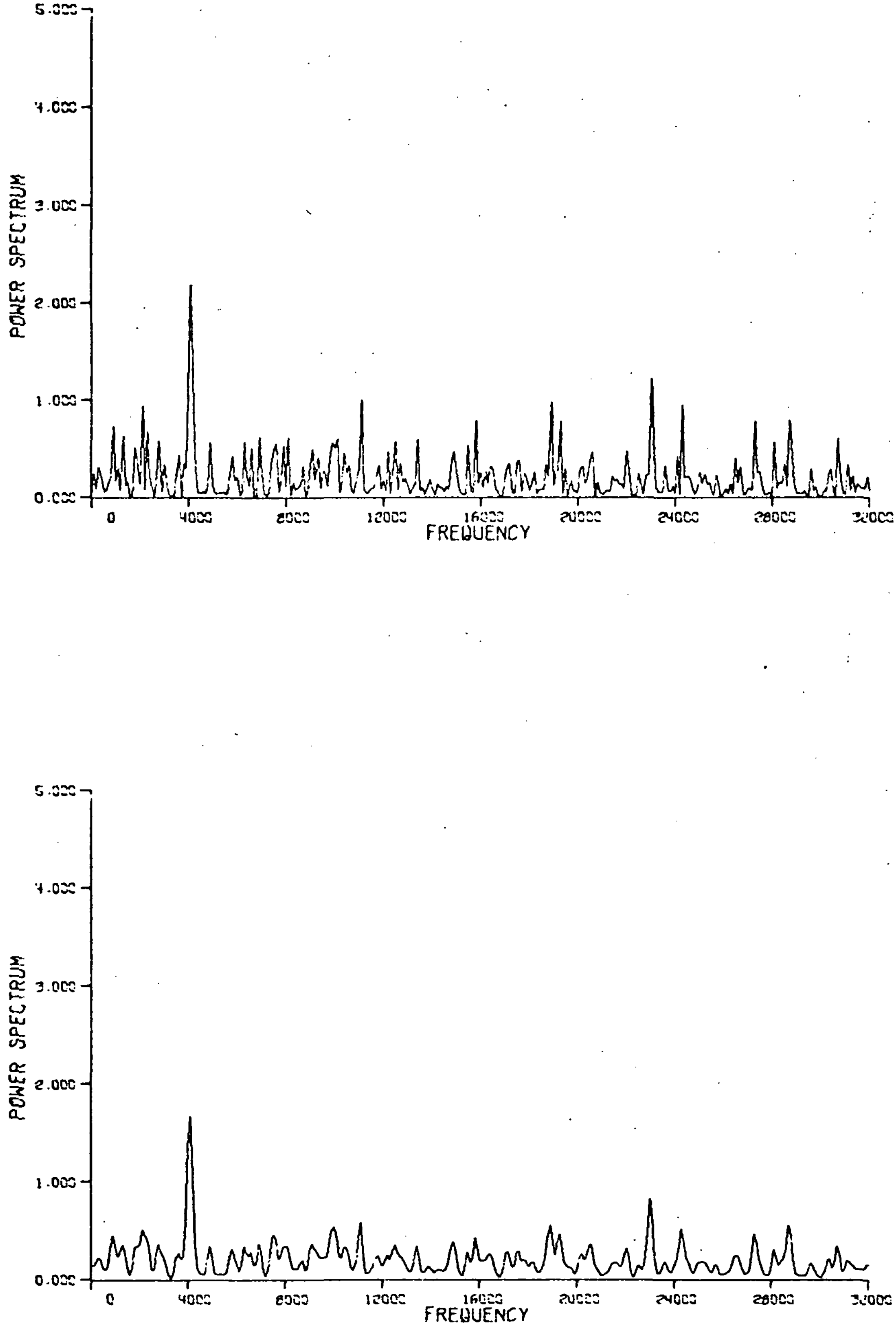

Fig. 3. Same signals as Fig. 1, but -15 dB signal-to-noise ratio.

2.2.3 Conclusions

The results presented in the preceding curves, and the more extensive calculations of a similar nature that have been performed, indicate a number of conclusions that can be drawn. Those conclusions that seem to be firmly established are:

a) Signals for which the signal-to-noise ratio (in the entire band) is greater than -14 dB are clearly detectable.

b) Signals for which the signal-to-noise ratio is less than -15 dB cannot be detected reliably.

c) Some smoothing of the spectral estimate is necessary for reliable detection, but the optimum trade-off between detection and resolution has not yet been established.

A tentative conclusion concerning the FFT is that it may be too time-consuming to allow the preamble to be shortened significantly. A preliminary estimate of the time required to compute the power spectrum from 1024 samples of data (gathered in 10 ms) and make the decisions as to the presence or absence of signals indicates that it would take about 40 ms with LSI logic or about 100 ms with IC logic. Hence, real-time computation is not possible and the number of complete computations that could be made in each signal would be limited.

2.3 FAST WALSH-HADAMARD TRANSFORM (FHT) METHODS OF COARSE FREQUENCY MEASUREMENT

2.3.1 Objectives

The Walsh basis functions are binary valued and can be described in terms of their zero crossings in a manner similar to that used for the sinusoidal basis functions of the Fourier transform. Hence, the average number of zero-crossings per second divided by two is the sequency and is related to the

frequency of sinusoids.

The objective in considering Walsh-Hadamard transforms is to reduce the computation time since each transform requires only $N \log_2 N$ real additions and no multiplications.

2.3.2 Discussion of Results

As in the case of Fourier methods it is desirable to consider the power spectrum (as a function of sequency) rather than the basic transform. Unlike the Fourier case, however, even the power spectrum is not invariant to time shifts. The discrete transform is defined as

$$B_n = \sum_{k=0}^{N-1} X_k \, \mathrm{wal}(n,k) \quad , \quad n = 0, 1, \ldots, N-1 \tag{2-11}$$

where wal(n,k) is the kth sample of the nth basis function. In this case n represents the number of zero-crossings in the interval on which the basis function is defined. The corresponding sequency power spectrum is

$$S(n/T) = \frac{T}{N} B_n^2 \tag{2-12}$$

The sequency power spectrum has been calculated for a large number of situations containing one and two signals at different frequencies and frequency separations. Some of these results are illustrated in Figs. 4 thru 7. All of these computations have been made for the case of <u>no</u> noise. No computations have been performed for the noisy case.

It is clear that in most cases maxima of the sequency power spectra do occur at the sequencies of the signal components. There are some unusual situations that arise, however. In all cases there are sub-maxima that occur on either side of the main peak. Usually these side lobes are smaller than the main peak but occasionally they can be larger. For an example of this, see Fig. 6.

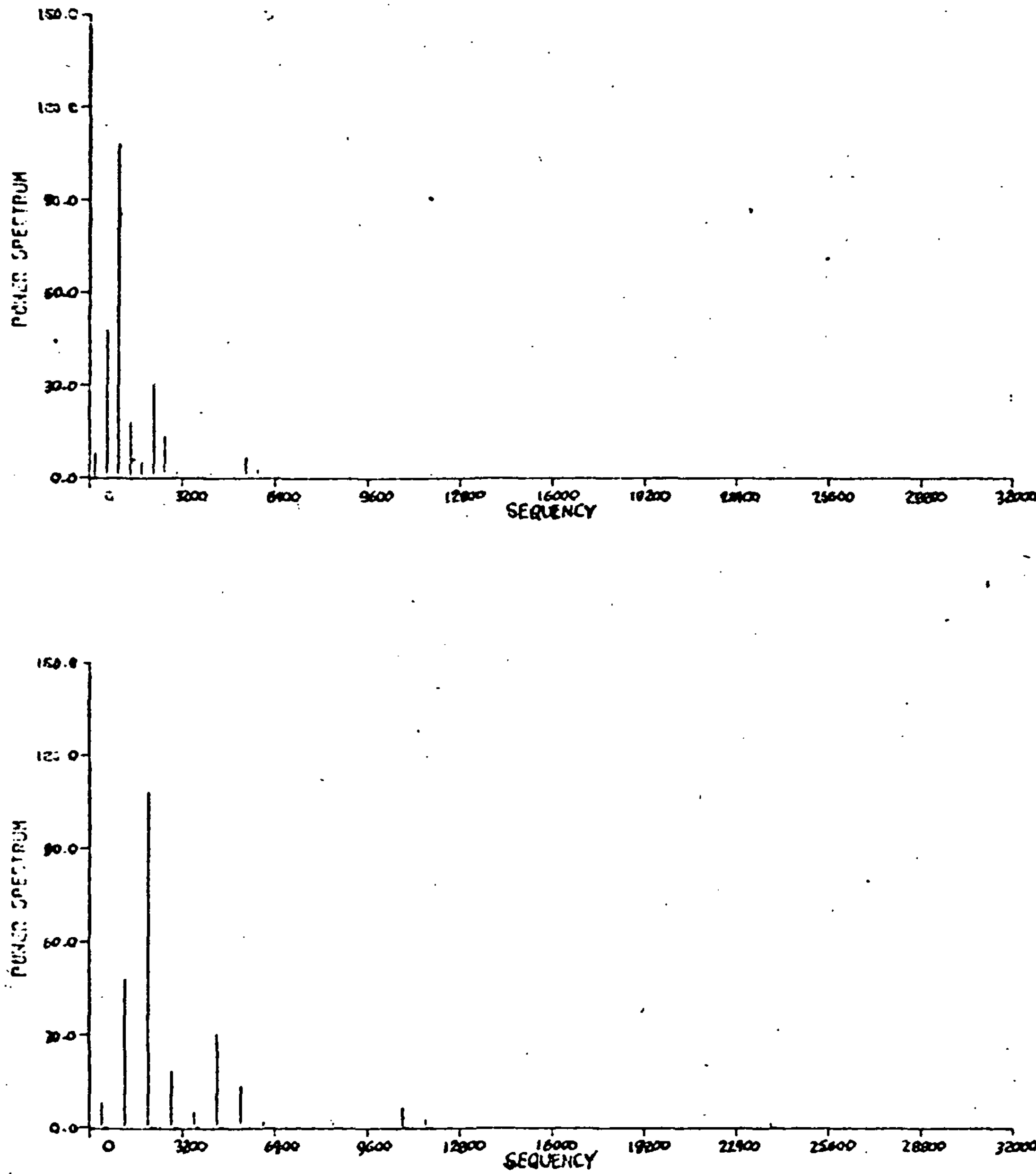

Fig. 4. Sequency power spectra. Upper plot: f = 1000 S_z^{Hz}.
Lower plot: f = 2000 S_z^{Hz}.

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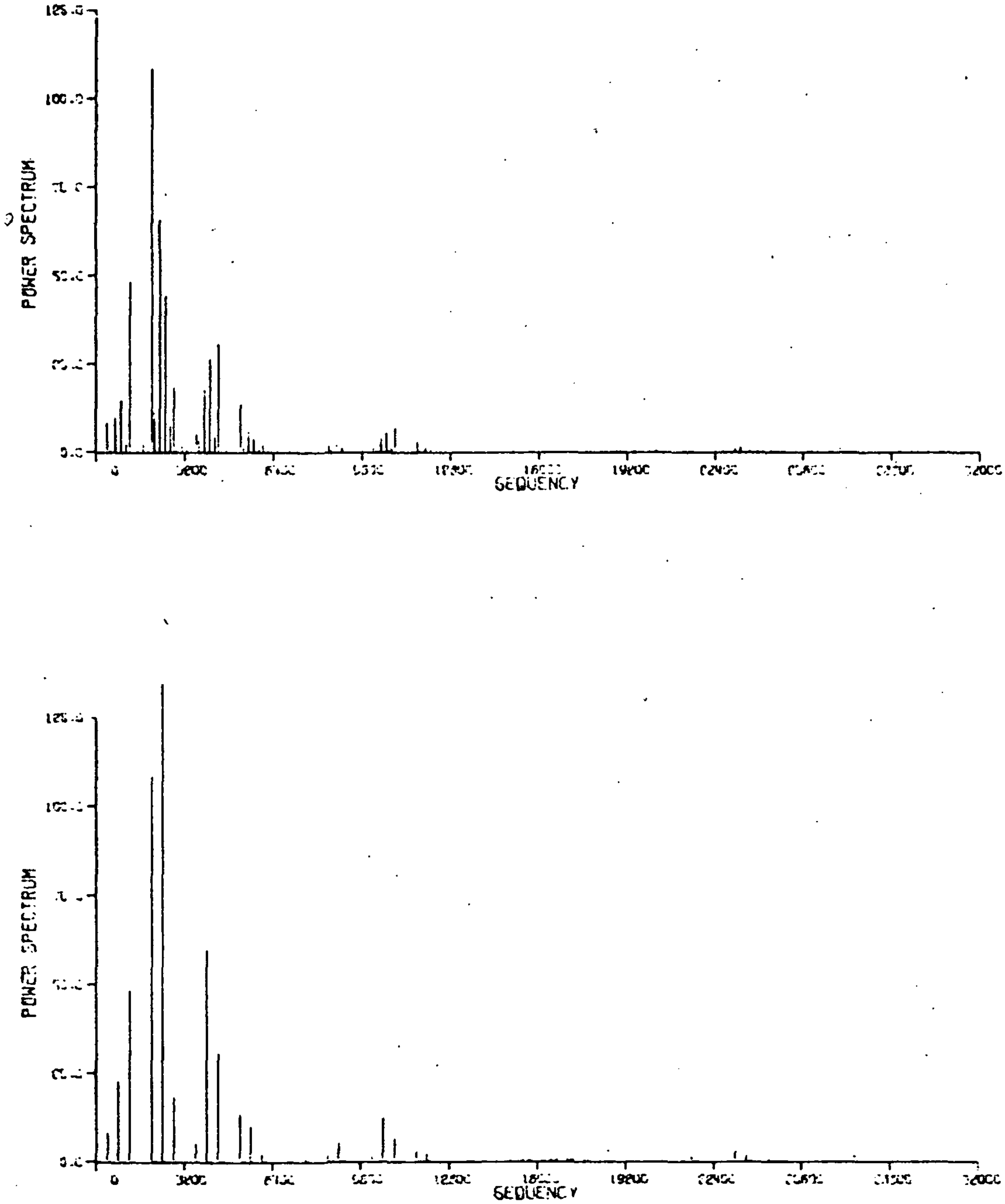
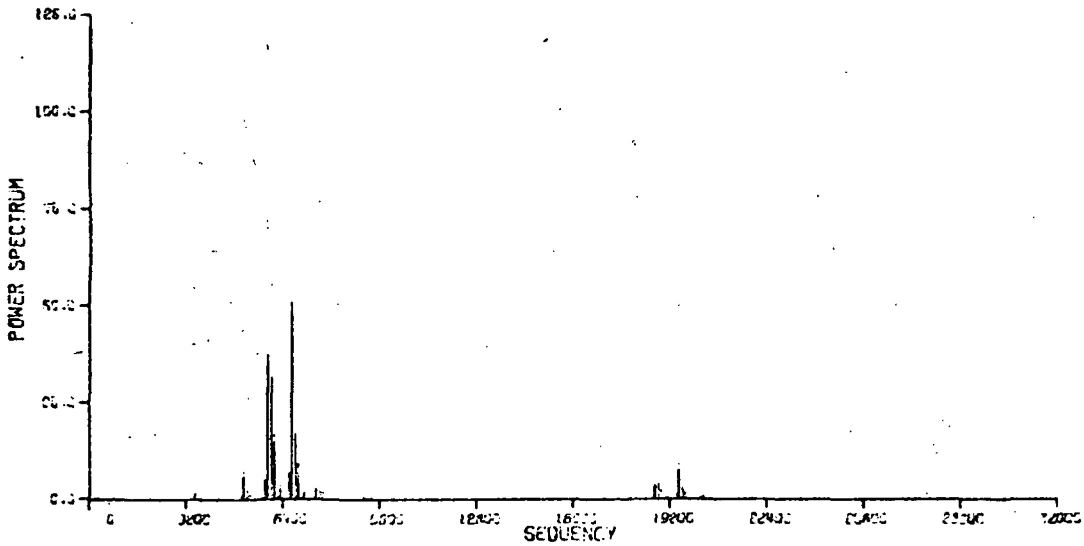


Fig. 5. Sequency power spectra of two equal amplitude signals. 20

Upper plot: $f_1 = 2000$ Hz, $f_2 = 2300$ Hz.

Lower plot: $f_1 = 2000$ Hz, $f_2 = 2400$ Hz.



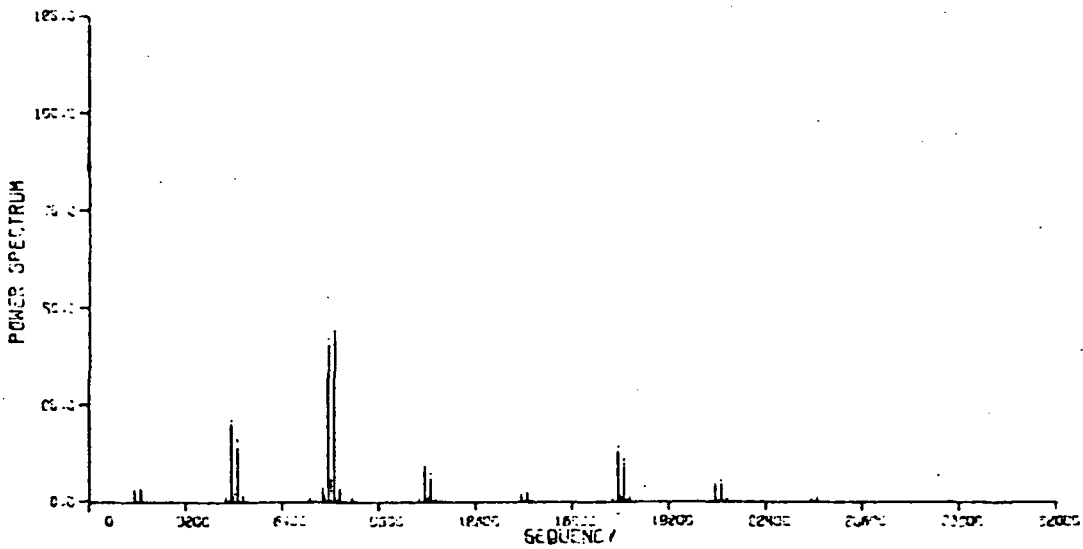


Fig. 6. Upper plot: Sequency power spectrum (f = 6060 Hz) for which the side lobe is larger than the main lobe. Lower plot: Sequency power spectrum for f = 8080 Hz.

There also appears to be an effect of weak signal suppression when one signal is much smaller than the other. This is illustrated in Fig. 7 in which one signal is 5 times larger than the other.

2.3.3 Conclusions

The only firm conclusion that can be reached at this time is that FHT methods are not thoroughly understood. On a more positive note, some tentative conclusions are:

a) So far as computation time is concerned, it is certainly less than the FFT, but how much less is not clear.

b) The desired sequencies can always be detected (in the noise-free case) but there may be false alarms introduced by the sidelobes.

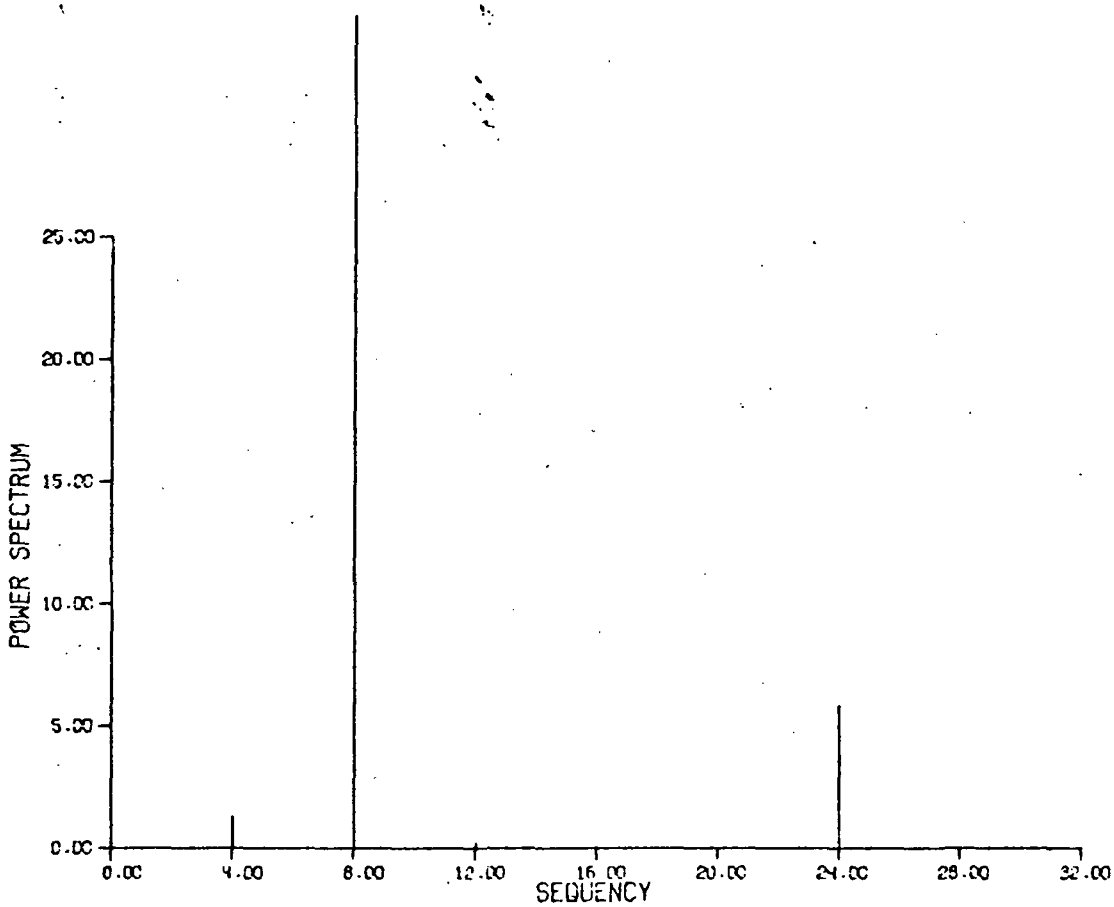
c) The apparent weak-signal suppression is probably an artifact of the particular computation rather than a fundamental limitation of the method. Like other transform methods, the FHT is linear and should not exhibit this effect in general, although it may for certain combinations of sequencies because of sidelobe interference.

It is clear that the available results for the FHT methods are still too fragmentary to be able to predict the eventual success of this technique. In particular, the effects of noise and the peculiarities of the sidelobes must be investigated further.

2.4 FILTER BANK METHODS OF COARSE FREQUENCY MEASUREMENT

2.4.1 Objectives

Another obvious method of detecting the presence of sinusoids of unknown frequency is the use of a bank of parallel filters tuned to contiguous frequency bands. For the parameters of the present study this would require 100 to 300 filters, depending upon the resolution required, and this number is deemed too



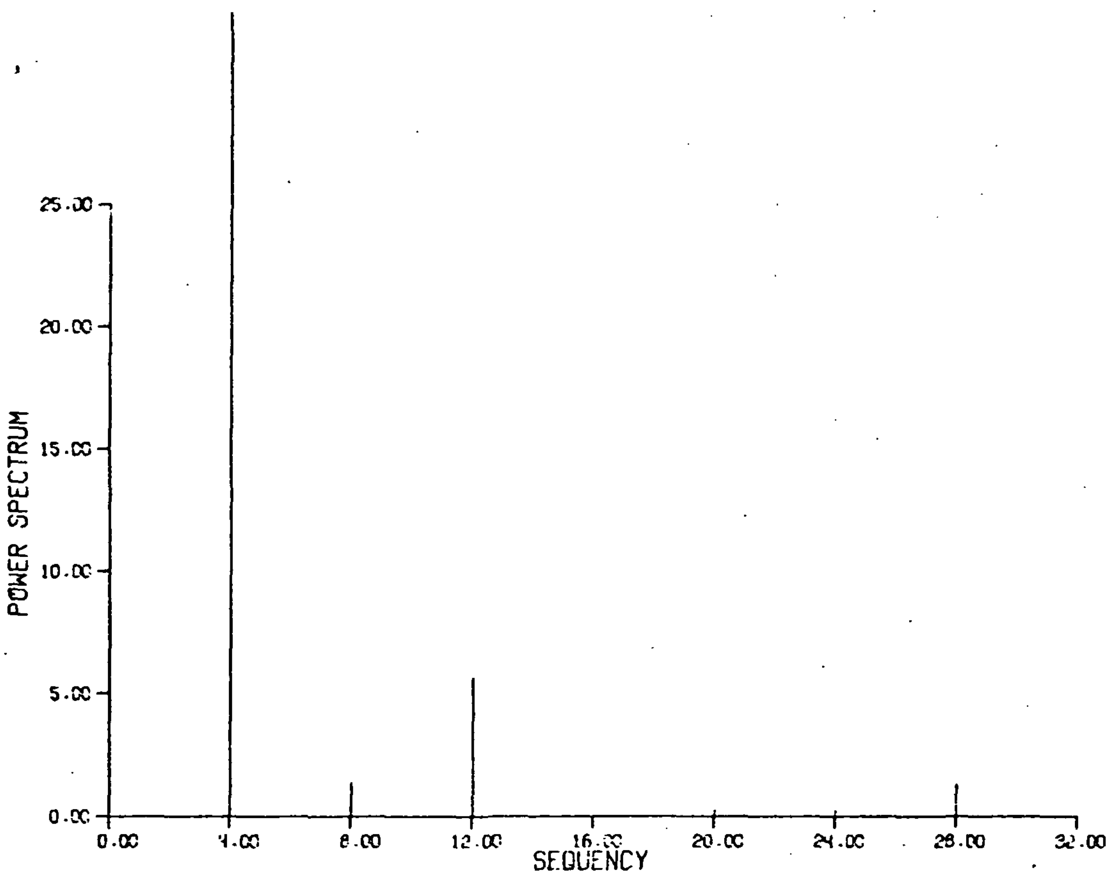


Fig. 7. Sequency power spectra of two signals (f_1 = 4.0, f_2 = 8.0) having a 25 to 1 power ratio.
Upper plot: f_2 = 8.0 is larger. Lower plot: f_1 = 4.0 is larger.

large for use of conventional filter structures. The alternative proposed was the use of digital filters, each scanning a pre-assigned portion of the overall frequency band.

The objective of this investigation is to determine the relative advantages and disadvantages of the digital filter bank with respect to processing time, memory requirements, and system complexity.

2.4.2 Discussion of Results

The investigation of digital filtering has been rather limited. Possible ways of implementing such a filter have been looked at and one possible technique is illustrated in Fig. 8. No analysis of its performance has been carried out.

2.4.3 Conclusions

Because of the lack of analysis, no firm conclusions regarding the feasibility of digital filtering have been reached. It has been tentatively concluded that the performance would be similar to the FFT but slower because there is no fast algorithm available and it is serial processor. Of course, savings in time can be achieved by using parallel processing of several bands simultaneously.

2.5 PHASE-LOCK LOOP FOR FINE FREQUENCY MEASUREMENT

2.5.1 Objectives

After the various frequency components of the received signal have been identified, it is necessary to lock on to each one in a separate sub-system in order to recover the information data. The phase-lock loop is proposed as the appropriate mechanism for acquiring the proper frequency and phase.

The objectives of this task are to determine the most appropriate configuration for the phase-lock loop and to evaluate its performance.

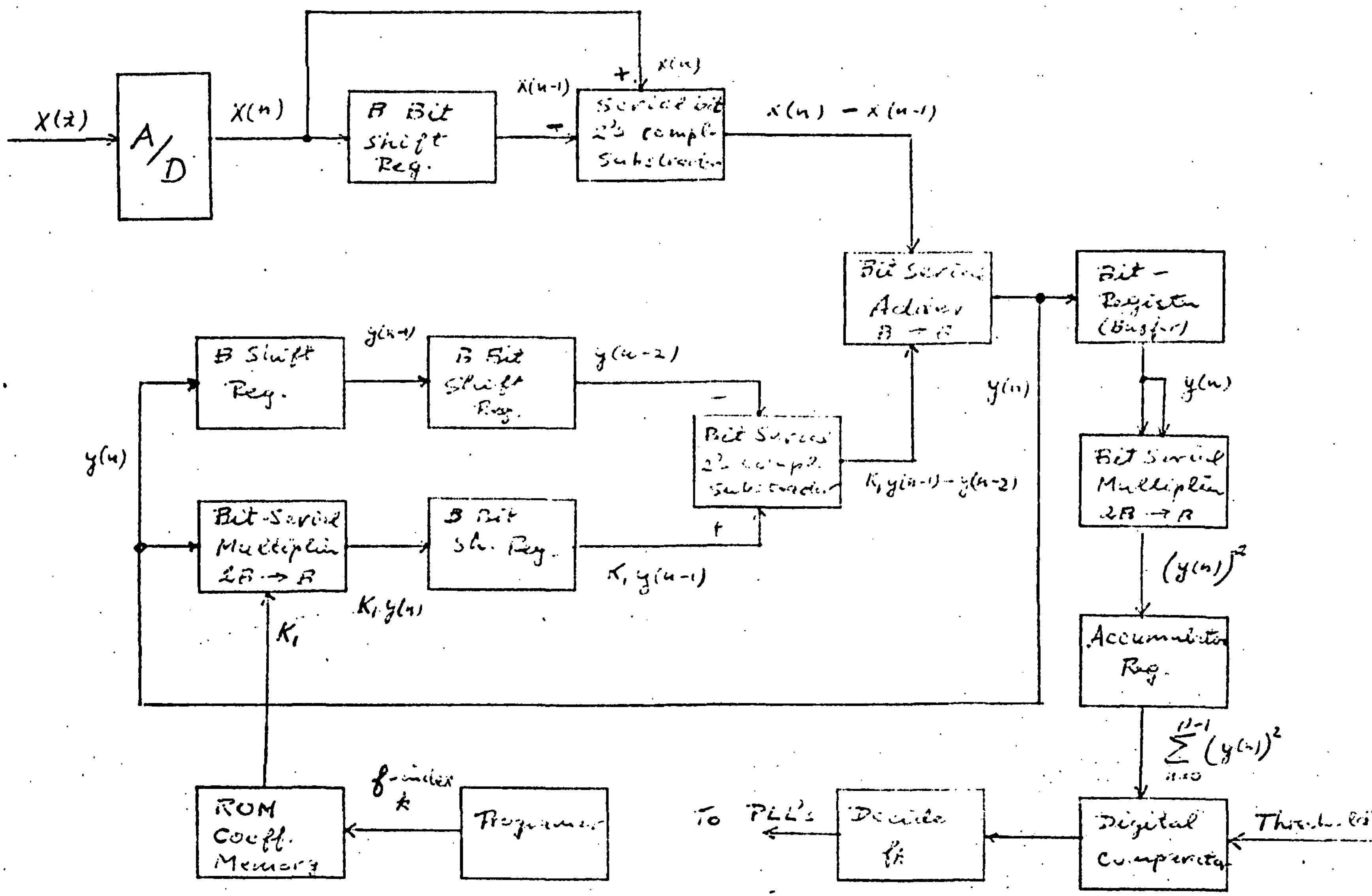


Fig. 8. Digital Resonator

2.5.2 Discussion of Results

Two different phase-lock loops have been considered. One of these is the squaring loop that removes the modulation (when it is PSK with $\pm 90^{\circ}$) by squaring the signal and operating at twice the carrier frequency. A block diagram of this loop is shown in Fig. 9.

The second method considered is the Costas loop that employs in-phase and quadrature feedback paths to remove the modulation (again assuming PSK with $\pm 90^{\circ}$). A block diagram of this loop is shown in Fig. 10.

The analysis of these phase-lock loops is only partially completed. The analysis techniques are the standard ones suggested by Stiffler and Lindsey, but no quantitative results have been obtained yet.

2.5.3 Conclusions

The only conclusion that has been reached so far concerning the fine frequency measurement is a tentative one that the performances of the two phase lock loops are essentially the same. If this turns out to be true, then the choice of method depends primarily upon the relative ease of implementing the two techniques.

2.6 ERRORS IN FREQUENCY IDENTIFICATION

2.6.1 Objectives

The coarse frequency measurements attempts to create peaks in the frequency spectrum or sequency spectrum of the observed data in order to identify the specific frequencies present. In the presence of noise, the correct peaks may be obscured or spurious peaks may be created. Hence, the objective of this task is to evaluate the effectiveness of a given coarse frequency measurement by establishing the probability of detection for the correct peaks and the probability of false alarm for the spurious peaks.

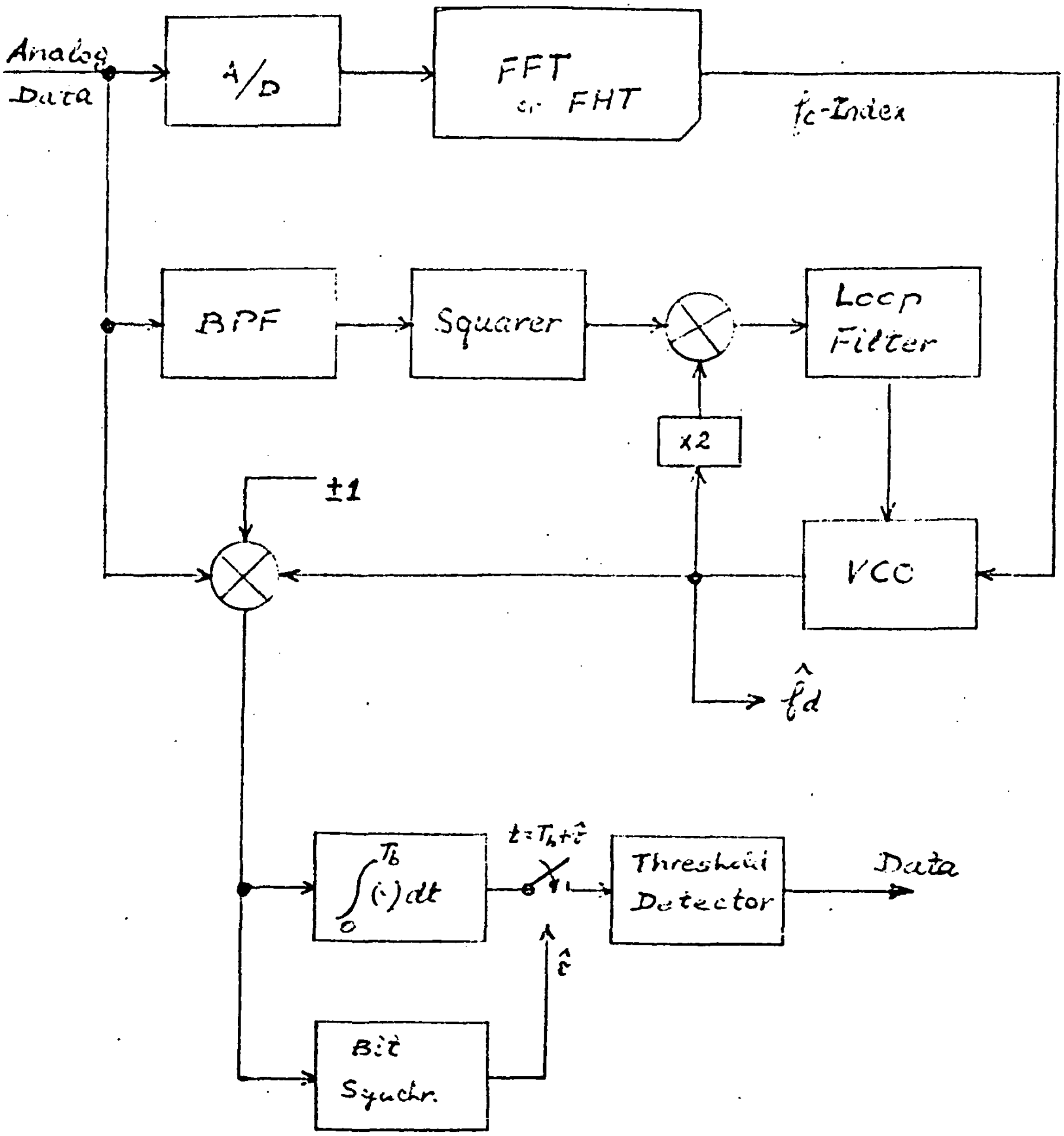


Fig. 9. Block diagram of the squaring phase-lock loop.

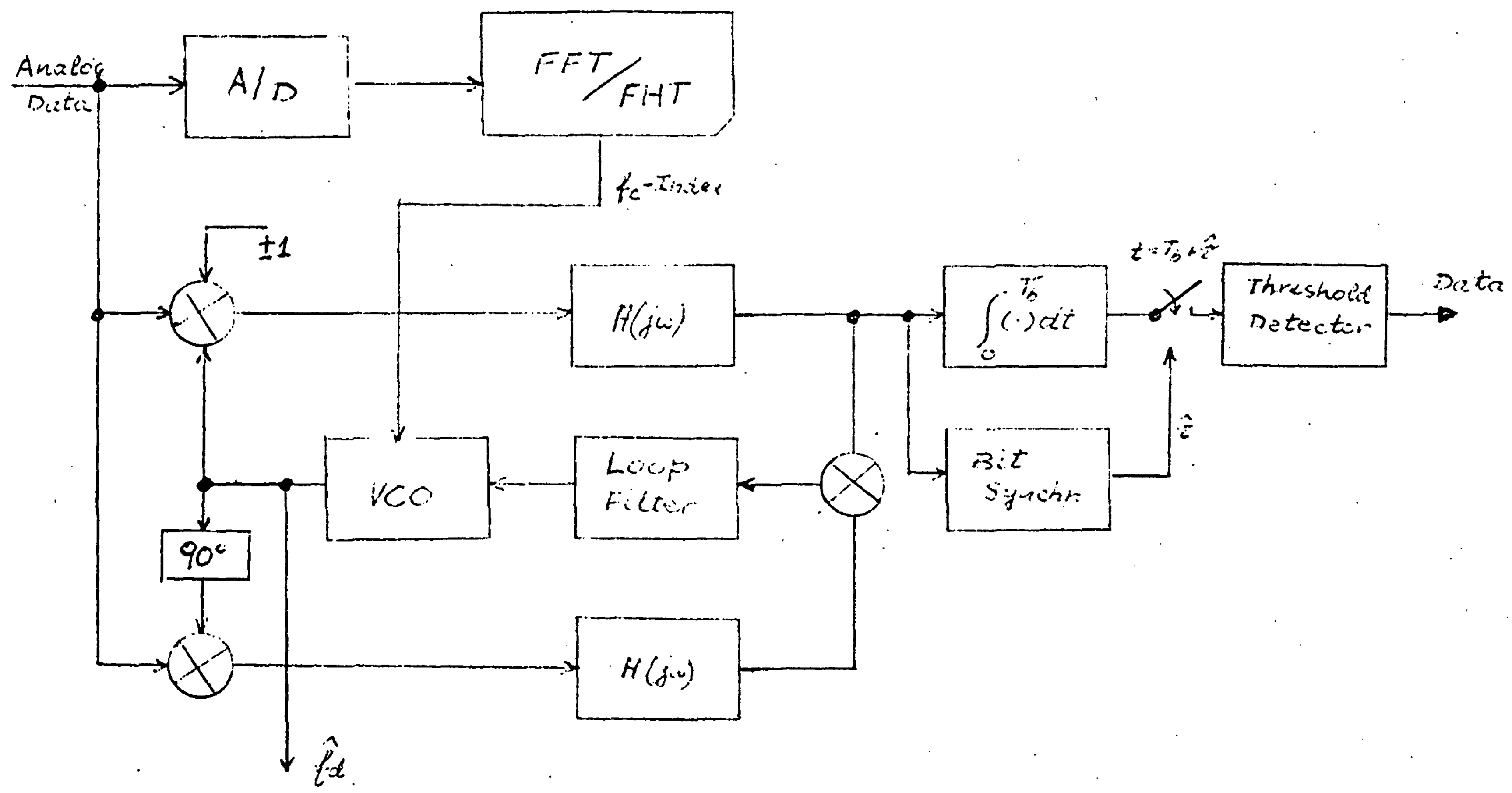


Fig. 10. Block diagram of the Costas phase-lock loop.

2.6.2 Discussion of Results

Only the FFT method has been investigated with respect to its error performance. The error analysis has been carried out by treating each estimate of the frequency spectrum as a random variable whose probability density function depends upon the magnitude of any signal components and their frequencies. A threshold is established and if the estimated spectrum lies above this threshold a decision of signal present is made.

The effectiveness of this decision operation can be expressed in terms of the probability of detection, P_D, and the probability of false alarm, P_F. The former is the probability that the spectral estimate of noise plus signal will exceed the threshold, while the latter is the probability that the spectral estimate of noise only will exceed the threshold.

The signal-to-noise ratio, R, of the spectral estimate is defined as the ratio of the expected value of the estimate when signal is present to the expected value of the estimate when only noise is present. Under suitable assumptions, the resulting probability density function is Chi-square with 2ξ degrees of freedom where ξ is determined by the smoothing as indicated in (2-8). Specifically it becomes

$$p_X(x;\xi;R) = e^{-R}\sum_{r=0}^{\infty} \frac{R^r\, x^{(r+\frac{\xi}{2}-1)}}{2^{(r+\frac{\xi}{2})}\,\Gamma(r+\frac{\xi}{2})\, r!}\, e^{-\frac{x}{2}} \qquad (2\text{-}13)$$

where $X = \dfrac{\xi|A_n|^2}{\mathrm{Var}(A_n)}$

and A_n is the Fourier coefficient defined in (2-1). This probability density function has been computed for numerous conditions and typical examples are shown in Figs. 11, 12, and 13 for various values of signal-to-noise ratio, R and with the degrees of freedom, ξ, as a parameter. In all cases the abscissa is proportional t

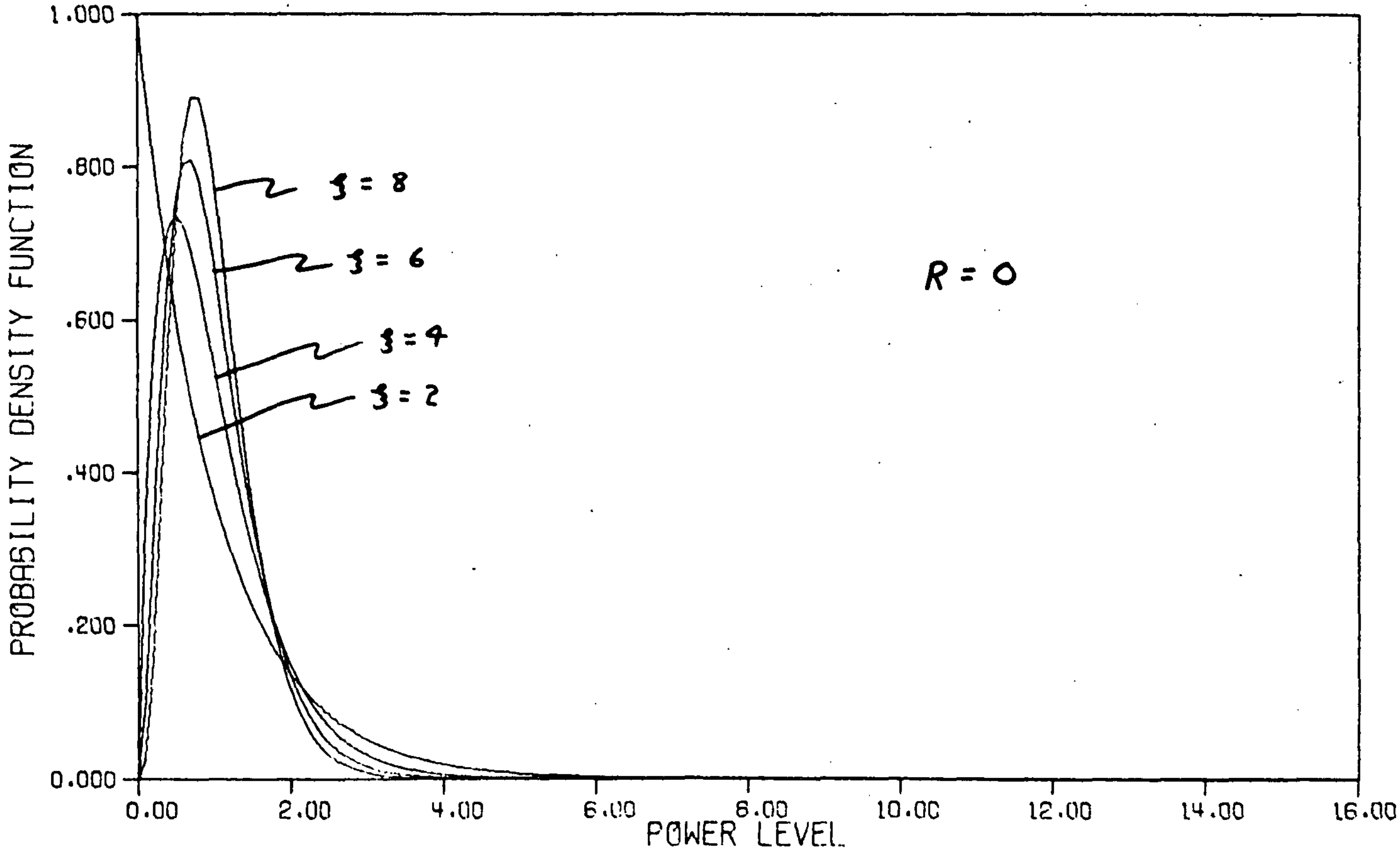


Fig. 11. Probability density function of spectral estimates of noise only.

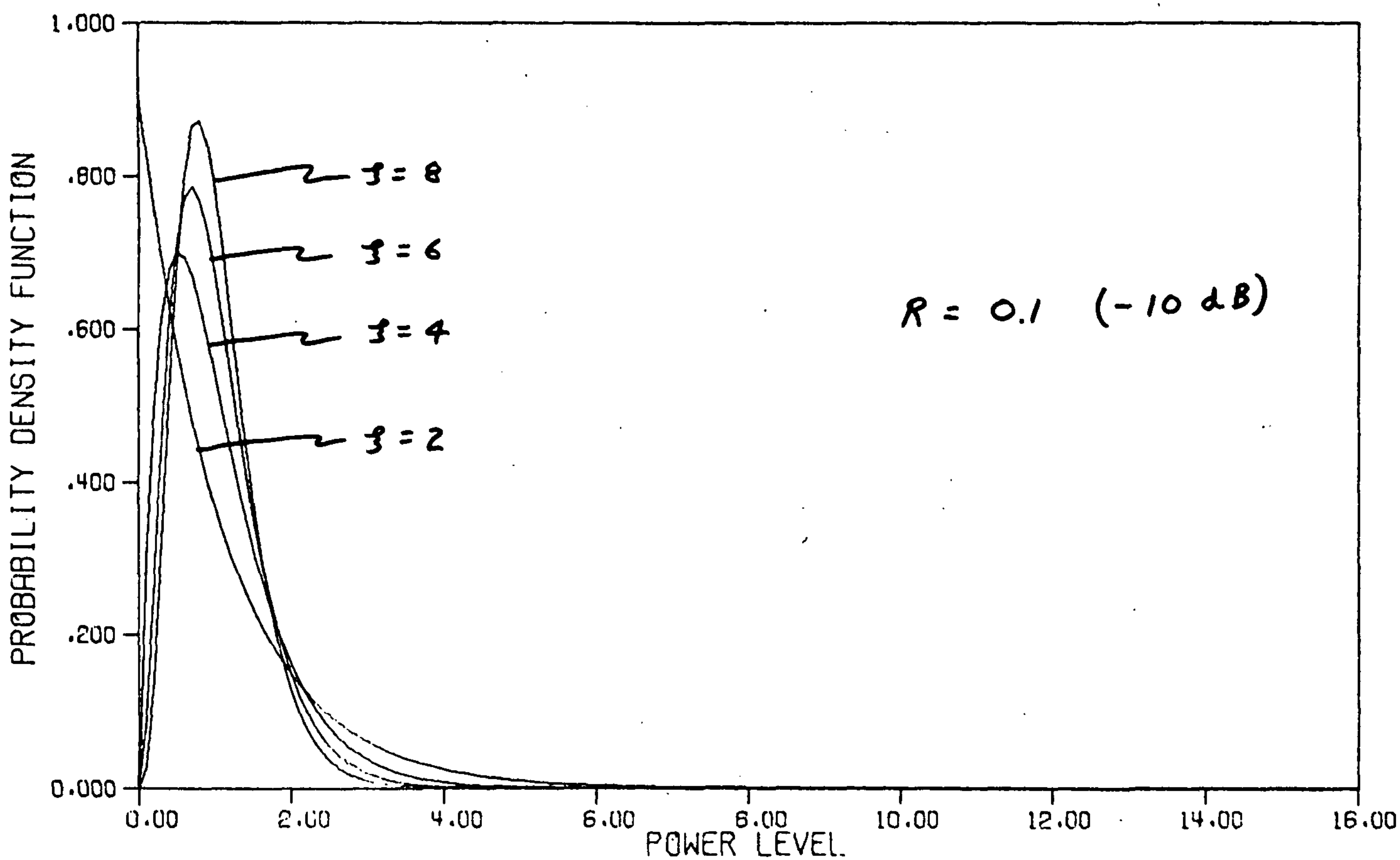


Fig. 12. Probability density function of spectral estimates for a weak signal.

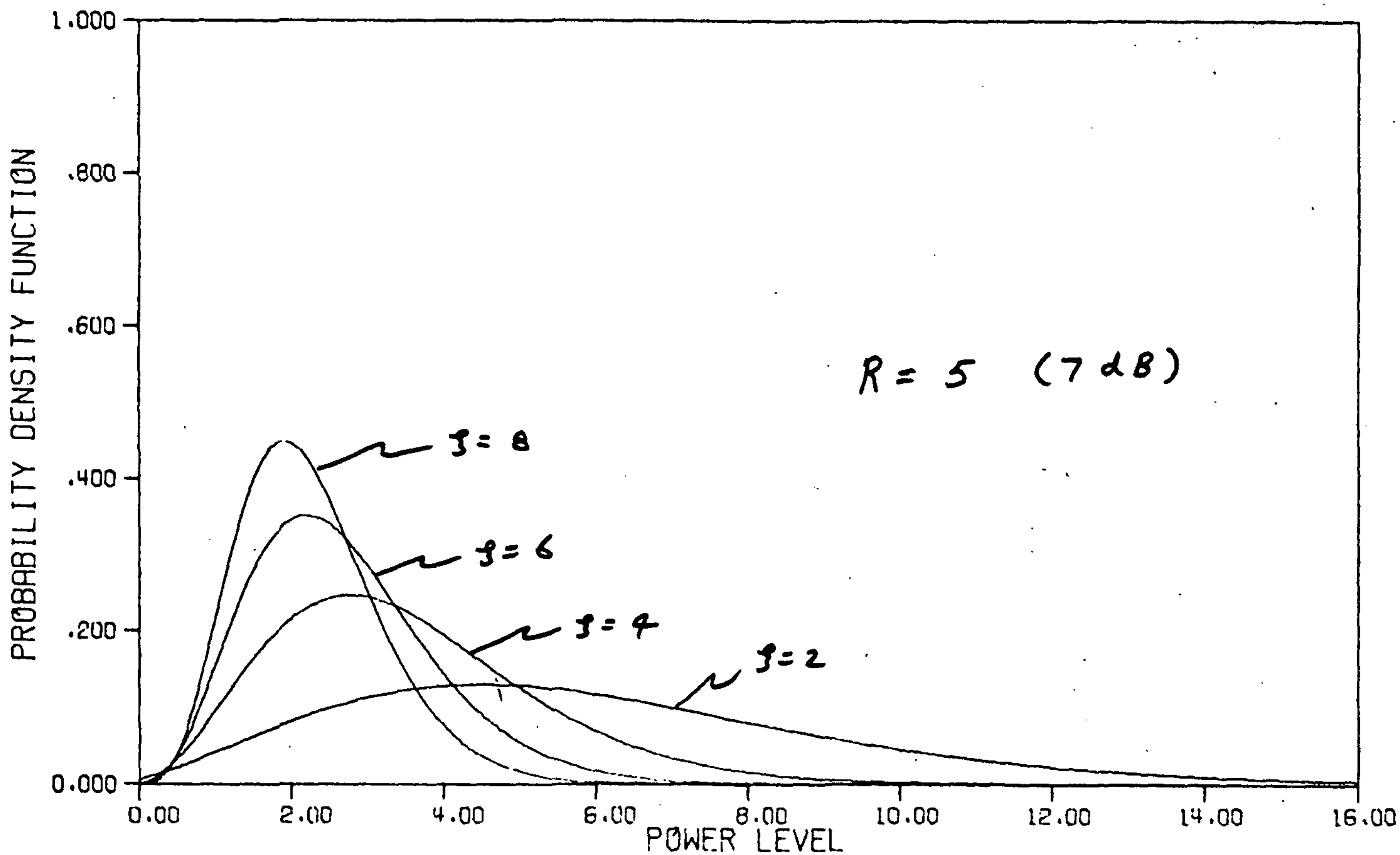


Fig. 13. Probability density function of spectral estimates for a strong signal.

the normalized power level, x.

Using the probability density function of (2-13) one can compute the probabilities of detection and false alarm for any signal-to-noise ratio and any number of degrees of freedom. If the decision threshold is set at $x = \delta$, then these probabilities become, for $\xi = 2$

$$P_D = e^{-(R+\delta)} \sum_{r=0}^{\infty} \frac{R^r}{r!} \sum_{i=0}^{r} \frac{\delta^i}{i!} \tag{2-14}$$

$$P_F = e^{-\delta} \tag{2-15}$$

One way of presenting the results of (2-14) and (2-15) is to plot curves of P_D as a function of R, with P_F as a parameter. One such set of curves, for 2 degrees of freedom, is shown in Fig. 14 for P_F values ranging from 10^{-1} to 10^{-7}.

Another way of presenting the same results is to plot curves of P_D as a function of R, with ξ as a parameter. This indicates how smoothing affects the probability of detection. One such set of curves, for $P_F = 10^{-4}$, is shown in Fig. 15.

Many computations similar to those of Figs. 11 thru 15 have been made, and the results presented are intended to be illustrative of these.

2.6.3 Conclusions

The conclusions that can be drawn as a result of this study pertain to Fourier transform methods only, since this is the only situation that has been investigated. Some fairly firm conclusions that can be made are:

a) It is a relatively easy matter to calculate P_D and P_F for a wide variety of cases when the FFT methods are used. Thus, the evaluation of this technique is straightforward.

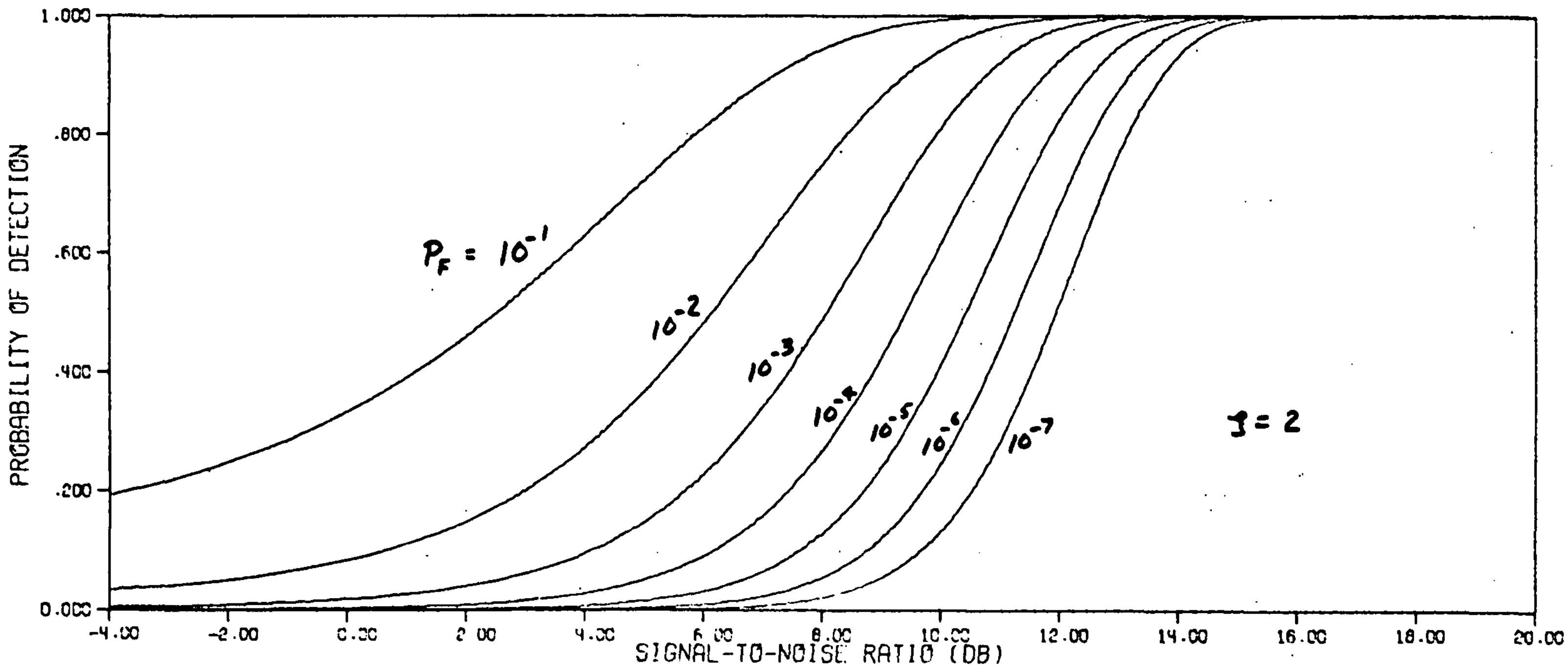


Fig. 14. Probability of detection with probability of false alarm as a parameter.

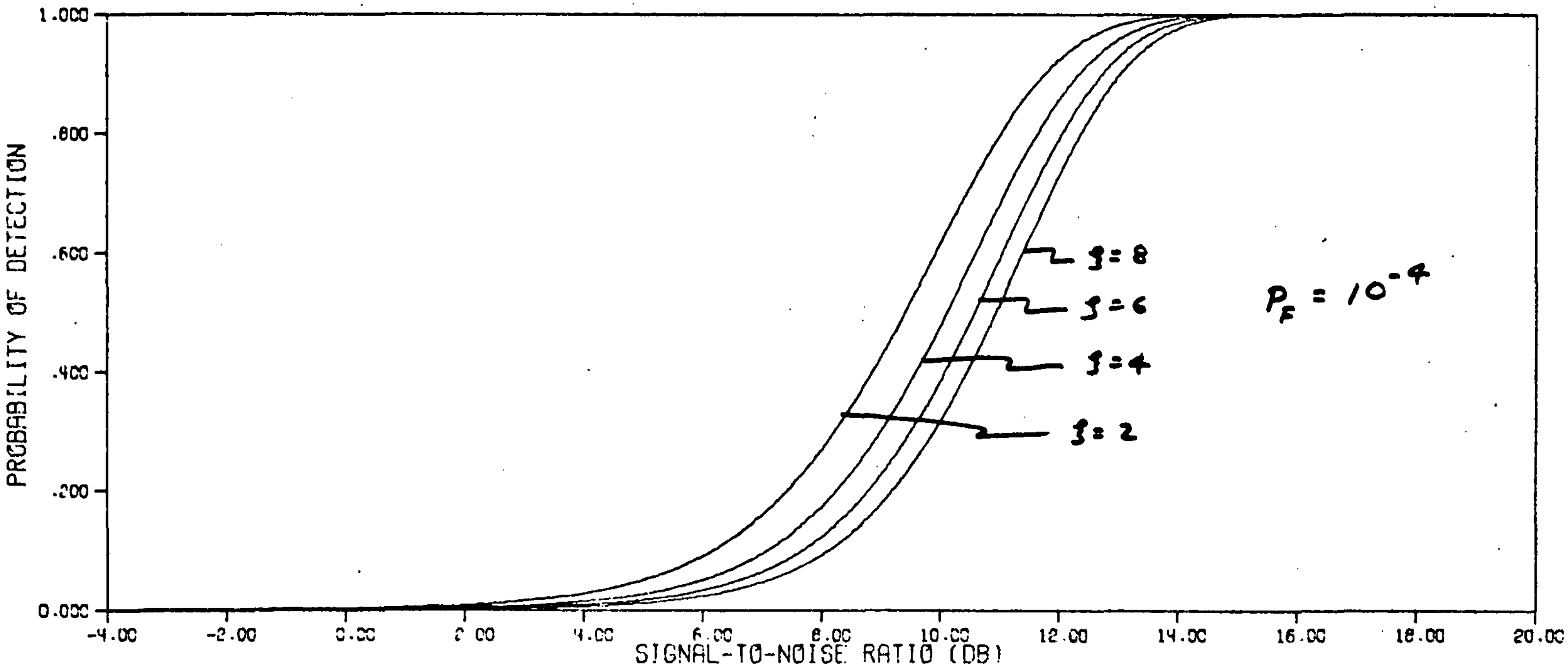


Fig. 15. Probability of detection with degrees of freedom as a parameter.

b) The values of P_D obtained by this computation are consistent with those that might be estimated from plots of the frequency power spectra.

c) As a typical reference point, it is seen from Fig. 14 that a signal-to-noise ratio of 10 dB (for the spectral estimate) leads to a probability of detection of 0.95 when there are 2 degrees of freedom (no smoothing). This corresponds to an input signal-to-noise ratio of about -12 dB.

2.7 BRIEF SUMMARY OF OTHER TASKS

2.7.1 Implementation of Coarse Frequency Measurement

Block diagrams of the hardware needed to accomplish the coarse frequency measurement by either the FFT or the FHT have been sketched. They are not presented here, however, because they are still tentative.

2.7.2 Use of Hard Limiting

It was suggested in the original proposal that the input signal and noise be hard limited before estimating the spectrum. This approach does not seem feasible, however, because of the suppression of weak signals and the creation of difference frequency components that would obscure the spectral estimates. No further work has been done on this idea.

2.7.3 Data Modulation and Demodulation

The use of PSK has been assumed throughout the analysis and no further consideration has been given to this problem. In particular, the trade-offs between PSK and DPSK have not been explored.

2.7.4 Message Format Design

The basic concepts of message format design were discussed in the original proposal. The mathematics for solving this optimization problem has been set-up but no solutions have been attempted. It should be noted, however, that the

solution of this problem will have an important bearing on the coarse frequency measurement.

Some work has been done on the synchronization problem, but this has not reached a sufficiently definitive stage to be worthy of reporting. No effort has been devoted to the suggestion that the preamble could be eliminated by storing the signal.

SECTION 3: RECOMMENDATIONS FOR FUTURE EFFORT

3.1 INTRODUCTION

If effort on this program is to be continued, it is essential to identify those tasks that are both worthwhile and feasible. The following discussion attempts to do this by considering tasks in three different categories:

a) Those tasks currently under investigation, or contained in the original proposal, that should be continued or initiated.

b) Those tasks in the original proposal that should be terminated or not initiated.

c) New tasks, some contained in the most recent proposal, that should be initiated.

The following subsections identify the above tasks, make specific recommendations, and briefly indicate the reasons for these recommendations.

3.2 CONTINUING TASKS

3.2.1 Fast Walsh-Hadamard Transform Methods of Coarse Frequency Measurement

The FHT methods have the advantage of being computationally efficient but there are many unanswered questions concerning their interpretation and use. These include:

a) The proper interpretation of the sequency spectrum for sinusoidal signals.

b) The significance of sidelobes on the sequency spectrum in creating false alarms.

c) The apparent weak signal suppression for some combinations of frequencies.

If the investigation of FHT methods is continued, it should concentrate on:

a) The possibility of controlling the sidelobes of the spectrum by the use

of appropriate weighting functions.

b) The effects of noise on the computation of sequency spectra, particularly with regard to establishing the minimum acceptable signal-to-noise ratio.

3.2.2 Digital Filter Bank

The investigation of the digital filter has not yet reached a satisfactory termination point. Although this approach does not look promising at the present time, this conclusion should be either verified or corrected.

3.2.3 Phase-Lock Loop

The analysis of the phase-lock loop for fine frequency measurement should be continued to the point of being able to specify performance and to select the most useful form of loop.

3.2.4 Message Format Design

It is essential to conclude the analysis of synchronization and to establish the optimum message format. However, this task need not be pursued in great depth.

3.2.5 System Error Analysis

The analysis of errors in frequency identification needs to be extended to other methods of coarse frequency measurement such as the FHT or some other method yet to be considered. This is an essential part of determining the relative effectiveness of the various methods.

Some analysis of data transmission errors is also needed but this is primarily an application of known techniques to the specific problem at hand. It is not envisioned that this will involve any extension of existing theory.

3.3 DISCONTINUED TASKS

3.3.1 Fast Fourier Transform Methods of Coarse Frequency Measurement

It appears that the analysis of the FFT has reached a point where it should be terminated, except for the final write-up and conclusions. It is more important at this time to bring the analysis of other methods of coarse frequency measurement to a similar level of completeness in order that final decisions on the approach can be made.

There may be some justification for looking at some modified FFT approaches, as indicated in a subsequent section.

3.3.2 Implementation of the Coarse Frequency Measurement

This task should be discontinued, at least until a final decision is made on the method to be used. However, since the evaluation of computation time is an important element in evaluating various methods, this aspect of the implementation must be continued.

3.3.3 Data Modulation and Demodulation

Since the assumption of PSK or DPSK seems to be well founded, there is little point in initiating activity in this area. Of course, if future effort indicates some definite advantages to be gained from large time-bandwidth waveforms and M-ary signalling, then this task would again become a crucial item.

3.3.4 Data Reduction

Although the eventual consideration of data reduction methods is vital to the design of the overall system, it does not appear to be an essential aspect of the detection and estimation problems that have been considered on this project so far. It is recommended that the effort on detection and estimation not be diluted by imposing a task on data reduction.

3.4 NEW TASKS

3.4.1 Theoretical Limitations

The fundamental problem of establishing limits on the accuracy of coarse frequency estimates has not been treated in any detail so far. A superficial look at the Cramér-Rao bound on such estimates indicates that the accuracy of estimating an isolated frequency component (in the presence of noise) is considerably better than the resolution obtainable with the FFT or FHT. This suggests that these are not efficient estimation techniques for isolated frequencies. The Cramér-Rao bound also can be evaluated for two or more closely spaced frequency components, but this has not been done. It may well turn out that the transform methods are not nearly as inefficient in this case.

It is suggested that these theoretical limits be investigated in order to provide an indication of how good practical estimation methods are. This is not a large task, but should provide much useful insight.

3.4.2 Adaptive Transform Methods

Two new methods of estimating power spectra have recently been proposed. These are the maximum likelihood method (MLM) proposed by Capon, and the maximum entropy method (MEM) proposed by Burg. More recently La Coss has demonstrated that these methods give substantially better resolution of sinusoidal signals in the presence of noise than do the classical methods.

It appears that some consideration should be given to the MLM and MEM, although it is likely that they are too time-consuming even though fast algorithms are available for both.

3.4.3 Nonlinear Processing

The knowledge that the signals to be detected are sinusoids can be utilized by employing nonlinear techniques to increase their separation in hertz. Such

nonlinear operations will also increase the noise but there may be a net improvement in resolvability. It appears that this possibility should be explored, at least to the extent of evaluating its potential.

Another approach to nonlinear processing is to examine the distribution of zero crossings of either the signal or its autocorrelation function. Such techniques have proven useful in weak signal detection in the past and might also be useful here. Unfortunately, there is very little relevant theory available on this problem so it would probably have to be attacked by a computer simulation.

There are undoubtedly many other nonlinear processing methods that could also be considered and some serious effort spent in contemplating some of these might be most fruitful.